STOCHASTIC COMPUTATION OF GAMMATONE FILTER BASED HEARING AID FOR IMPAIRED PEOPLE

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Abstract - *For the people with disabilities, information provided through their own telecommunication devices is indispensable to move. We propose this wireless system to send information to people with disabilities (especially deaf people).The most effective way to compensate hearing loss is to employ a hearing aid system which is an integration of voice amplification, noise reduction, feedback suppression, automatic program switching, environmental adaptation, etc. This project introduces about gammatone filter based upon stochastic computation for area efficient hardware. The gammatone filter well expresses the performance of human auditory peripheral mechanism. This filter has a potential of improving advanced speech communications systems, especially hearing assisting devices and noise robust speech recognition systems. Using the gain-balancing techniques, the computation accuracy at each IIR filter is improved, leading to a high dynamic range.*

Key Words: Gammatone filter, IIR filter, Gain-balancing techniques.

1. INTRODUCTION

There are two types of hearing loss, conductive and sensorineural hearing loss. For conductive hearing loss, there are different types of hearing aids available whereas for sensorineural hearing loss there is only surgery available that is Bone Anchored Hearing Aid. Since this surgery is done by drilling the bone near the ear. This surgery is costly and leads to uneasiness in speaking and while eating. This causes infection and need for surgery again and again. To overcome this surgery we use gammatone filter based hearing aid.

Using gammatone filter enhances the dynamic range, the gain balancing technique provided splits the original small gain into larger multiple gain. Gammatone filter in voice IC is used for filtering and amplifying. Here the microcontroller is used for controlling process, when it detects a sound, it activates the voice IC and it starts to record.

Voice IC is used to regenerator the sound after microcontroller's signal. This signal is converted to the required form using DC motors used in toy cars and it is given to the auditory nerves present in the teeth. This is cost efficient and can be used easily by people.

2. VOICE IC

Voice IC single chip used to record and playback. When the microcontroller detects the sound by sound sensor, sends the signal for recording. The Microphone present in the voice is records the sound signal. It converts the sound waves into audio signal. This audio signal which is in the form of analog signal is filtered and converted into digital signal. This signal is compressed and stored in memory. This process takes place till the microcontroller detects the sound. After the few seconds of delay in the sound, the microcontroller sends the signal to playback. This signal is in digital form, cannot be supplied to motor for our purpose. So it has to be converted to analog which is done by Pulse Width Modulated (PWM) signal generated by the microcontroller.

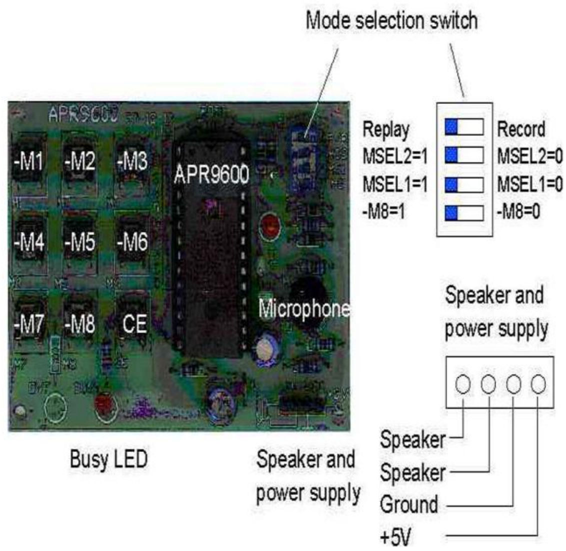

Fig 1: Voice IC Module

This voice IC work in two modes, serial and parallel mode. We use serial mode in which recording and replaying can be done in following way

2.1 SERIAL MODE RECORDING AND REPLAYING

2.1.1 Record sound tracks sequentially

This is an example of recording sequential sound tracks. The mode switch should have the following pattern: MSEL1=0(switched to right-hand side of the mode selection switch), MSEL2=0 (right-hand side). –M8=1 (left-hand side). RE=0 (right-hand side). Press CE first to reset the sound track counter to zero. Press and hold –M1 down and you will see BUZY LED illuminates. You can now speak to the microphone. Recording will terminate if –M1 is released or if the recording time exceeds 60 seconds (in this case you will run out the memory for your next sound track). Press –M1 again and again to record 2nd, 3rd, 4th and other consecutive sound tracks. Each sound track may have different lengths, but the accumulated length of all sound tracks will not exceed 60 seconds.

2.1.2 Replay sound tracks sequentially

Now make RE=1 (switched to Left-hand side of the mode selection switch) while keep other switches at the same location. Toggle –M1 (press key and release) causes the 1st sound track to be played once. Toggle –M1 again and again will play the 2nd, 3rd, 4th and other consecutive sound tracks. Press CE to reset the sound track counter to zero.

2.1.3 Record sound tracks with forward control

This is an example of recording sound tracks with forward control. The mode switch should have the following pattern: MSEL1=0(switched to right-hand side of the mode selection switch), MSEL2=0 (right-hand side). –M8=0 (right-hand side). RE=0 (right-hand side). Press CE first to reset the sound track counter to zero. This mode is rather similar to the above sequential sound recording. The only difference is that after –M1 is pressed and released, the sound track counter does not increment itself to the next sound track location. To move to the next sound track, –M2 should be toggled. So if –M1 is not toggled again and again without toggling –M2, sound will be recorded at the same sound track location.

2.1.4 Replay sound tracks with forward control

Now make RE=1 (switched to Left-hand side of the mode selection switch) while keep other switches at the same location. Toggle –M1 (press key and release) causes the 1st sound track to be played once. Toggle –M1 again and again will still play the 1st sound track. Once –M2 is toggled, the sound track counter is incremented and the next sound can be played. Press CE to reset the sound track counter to zero.

3. GAMMATONE FILTER

For brainwave auditory signal processing, a gammatone filter that has a similar response to the impulse responses of basilar membrane is a promising technique for advanced speech communications systems, such as cochlear implants and noise robust speech recognitions. The gammatone filter requires a high computational power because its function is complex.

Gammatone filter can be implemented by using two implementation techniques namely analog and digital implementation. Even though analog implementation is a beneficial in low power consumption and low hardware area, it experiences process variations.

In digital implementation the gammatone filter is designed as using multipliers of IIR (Infinite Impulse Response) filters is cascaded in series. The design method follows a stochastic computation technique. This technique represents data in stream of random bits.

When a voice signal enters the gammatone filter it changes the signal from frequency domain to time domain. Then the signal is segmented and correspondingly the amplification factor increases and noise is filtered. Noise is filtered by the filter bank of the gammatone filter known as Equal Ratio filter Bank (ERB).

The amplification factor is increased by improving the gain in each IIR filter by giving a small amount of threshold gain in each filter (soft threshold) thus by increasing the total gain of filter the dynamic range of filter is improved. The gammatone filter is designed to achieve a gain of 71.71 dB and other filters can only filter up to 5.74dB.

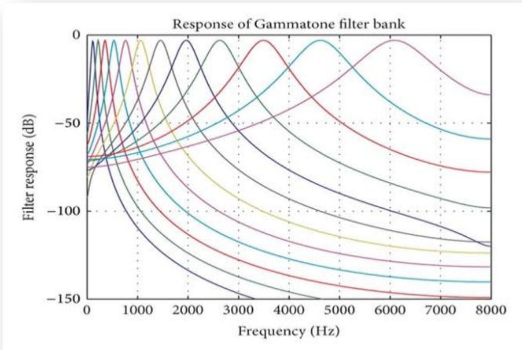

Fig -2: Frequency responses of gammatone filter.

3.1 Transformation of system

A gammatone filter is represented by an impulse response that is the product of a gamma distribution and a sinusoidal tone as follows

$$g(t) = at^{n-1}e^{-2\pi bERB(fc)t}\cos(2\pi f_c t + \varphi)\ (t > 0), \quad (1)$$

where a is a constant, n is the order of the filter, b is the bandwidth of the filter, fc is the center frequency of the filter, and φ is the steering phase. The equation can represent the human auditory filter when n is 4 and is 1.019 times Equivalent Rectangular Bandwidth (ERB). The ERB can be approximated as follows

$$ERB(f_c) = 24.7(4.37f_c/1000 + 1) \quad (2)$$

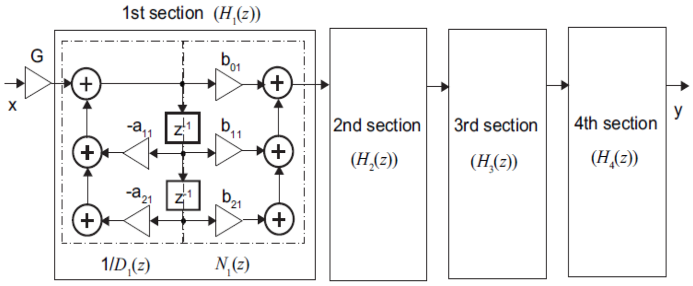

Fig -3: Block diagram of four cascaded second-order sections for gammatone filters.

In this paper, a is set to 1 and φ is set to 0 as. The frequency responses of the gammatone filters are shown in Fig. 2, where fc are 0.5, 1, 2, 5, and 10 kHz. The gammatone impulse response is converted to that in the frequency domain using the Laplace transform, which is then converted to a digital IIR filter using the bilinear transform with a fc of 5 kHz and a sampling frequency fs of 20 kHz used in this paper. The transfer function in digital domain H(z), is described using an 8th-order digital IIR filter as follows

$$H(z) = \frac{b_0 + b_1z^{-1} + \ldots + b_8z^{-8}}{1 + a_1z^{-1} + \ldots + a_8z^{-8}}, \quad (3)$$

where b_n $(0 \le n \le 8)$ and a_m $(1 \le m \le 8)$ are coefficients.

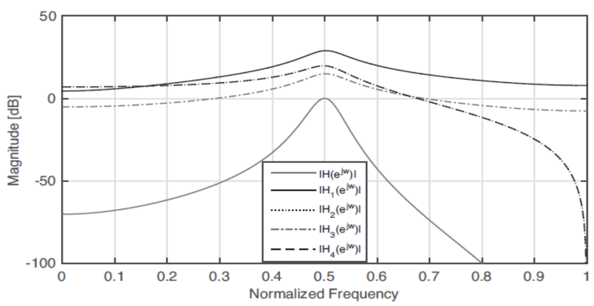

Fig -4: Magnitudes of frequency responses in the four cascaded second-order IIR filter for a gammatone response, where the center frequency is 0.5π rad.

3.2 Stochastic Implementation of Gammatone Filter

Stochastic computation has been recently exploited for several applications, such as low-density parity-code (LDPC) decoding, image processing, and digital filters. In stochastic computation, information is carried by the frequency of ones in a sequence in one of two formats: *unipolar* and *bipolar* coding. Note the probability of observing a '1' to be Pa =Pr(a(t) = 1) for a sequence of bits a(t). A value a is a = Pa, (0 ≤ a ≤ 1) in *unipolar* coding and is a = (2 ・ Pa − 1), (−1 ≤ a ≤ 1) in *bipolar* coding.

A multiplier is simply designed using a simple logic gate, such as a 2-input AND gate in unipolar coding or a 2- input XNOR gate in bipolar coding shown in Fig. 4 (a) and (b). The output probability, Pc, is (Pa ・ Pb) in unipolar coding. In the example shown in Fig. 5 (a), input values are represented using 10 bits and are multiplied with 10 clock cycles. Fig.4 (c) shows a block diagram of a two-input scaled addition designed using a two-input multiplexor, unlike a binary full adder. The output probability, Pc, is Ps ・ (Pa +Pb), where Ps is a probability of a selector input to the multiplexor.

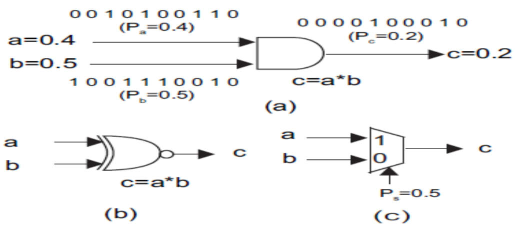

Fig -5: Stochastic computation blocks: (a) multiplier in unipolar coding, (b) multiplier in bipolar coding, and (c) scaled adder.

4. DC MOTOR

In any electric motor, operation is based on simple electromagnetism. A current-carrying conductor generates a magnetic field when this is then placed in an external magnetic field, it will experience a force proportional to the current in the conductor, and to the strength of the external magnetic field. . The internal configuration of a DC motor is designed to harness the magnetic interaction between a current-carrying conductor and an external magnetic field to generate rotational motion.

The shunt motor is different from the series motor in that the field winding is connected in parallel with the armature instead of in series. Since the field winding is placed in parallel with the armature, it is called a shunt winding and the motor is called a shunt motor. the field terminals are marked Fl and F2, and the armature terminals are marked Al and A2.

Every DC motor has six basic parts -- axle, rotor, stator, commutator, field magnet(s), and brushes. In most common DC motors (and all that Beamers will see), the external magnetic field is produced by high-strength permanent magnets[1]. The stator is the stationary part of the motor, this includes the motor casing, as well as two or more permanent magnet pole pieces. The rotor (together with the axle and attached commutator) rotates with respect to the stator. The rotor consists of windings (generally on a core), the windings being electrically connected to the commutator.

In this paper the output from the voice board, which is in the form of analog signal is amplified and given the DC motor. When the electric current passes through the coil of the motor, causes the shaft to vibrate.

When this vibration is directed to the teeth, which has the auditory nerves. The vibration stimulates the nerve endings in the cochlea which send nerve impulse to the brain via auditory nerve are then interpreted as sound.

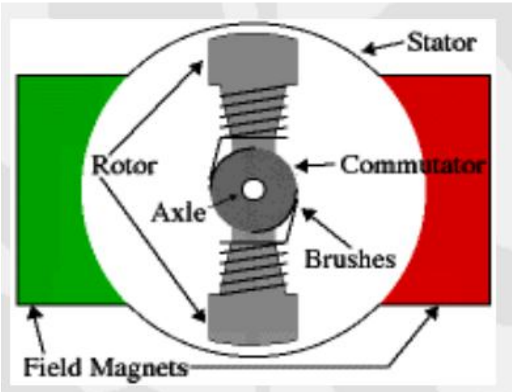

Fig-6: Brushless DC motor

5. CONCLUSION

A unique technological approach for the treatment of sensorineural hearing loss. Unlike BAHA, it does not require surgery. This device removes the need for a surgical implant in skull which eliminates the costs and complications of surgery.

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